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# Interaction SIP Bulk Caller

## Technical Reference

### Abstract

This document shows how to install, configure, and use Interaction SIP Bulk Caller, a utility that enables you to test your server by making or receiving (in bulk) large numbers of calls. It also simulates station registrations for testing.

For the latest version of this document, see the PureConnect Documentation Library at: <http://help.genesys.com/pureconnect>.

For copyright and trademark information, see [https://help.genesys.com/pureconnect/desktop/copyright\\_and\\_trademark\\_information.htm](https://help.genesys.com/pureconnect/desktop/copyright_and_trademark_information.htm).

# Table of Contents

Table of Contents	2
SIP Bulk Caller Overview	3
System Requirements	3
Glossary	3
Installing Interaction SIP Bulk Caller	4
Upgrading Interaction SIP Bulk Caller	4
Uninstalling Interaction SIP Bulk Caller	4
Installing Interaction SIP Bulk Caller software	4
Applying the License	6
Configuring and Using the Software	8
Adding a Callset or Stationset	8
Editing, Reloading, or Deleting a Callset or Stationset	9
Edit SIP Line Configuration	10
Edit Audio Configuration	12
Start or Stop Callset or Stationset	14
Generating Security Certificates for TLS	18
Location of Log Files	18
Troubleshooting	18
Appendix: Configuration Files	19
Outbound Callset Configuration File	20
Explanation of Outbound Callset Configuration File	20
Inbound Callset Configuration File	23
Explanation of Inbound Callset Configuration File	23
StationSet Configuration File	25
Explanation of StationSet Configuration File	25
Main Xml Configuration File	26
Explanation of Main Xml Configuration File	27
Registering a station based on a SIP name	27
Example	27
Change Log	28

# SIP Bulk Caller Overview

Interaction SIP Bulk Caller is a utility that enables you to test your CIC server by making or receiving (in bulk) large numbers of calls. It also simulates station registrations for testing.

Though it is normally used with a CIC server, Interaction SIP Bulk Caller does not require it and can be used with other servers.

## System Requirements

Interaction SIP Bulk Caller requires:

- Windows 7 or higher version
- MSXML 6.0 Parser

**Note:** Do not install Interaction SIP Bulk Caller on the same machine where SIP Proxy, CIC server, or any CIC server-related component (which uses the SIP Engine) is installed.

## Glossary

**\$SBC\_ROOT:** The folder where you installed Interaction SIP Bulk Caller.

**Callset:** A callset is a configuration and set of actions to use for each call. The callset XML configuration file describes it. There are two types of callsets:

- Use an outbound callset to place outbound calls to remote machines. Outbound callset is listed in the OutboundConfig section of the main XML configuration file.
- Use an inbound callset to respond to calls received by Interaction SIP Bulk Caller. Inbound callset is listed in the InboundConfig section of the main XML configuration file.

**SIPBulkCaller:** Interaction SIP Bulk Caller works both as an outbound caller and as an inbound receiver. When used to make calls, it is often referred to as SIPBulkCaller. When used to receive calls, it is often referred to as SIPBulkReceiver.

**SIPBulkReceiver:** Interaction SIP Bulk Caller works both as an outbound caller and as an inbound receiver. When used to make calls, it is often referred to as SIPBulkCaller. When used to receive calls, it is often referred to as SIPBulkReceiver.

**Stationset:** A stationset is a set of stations that Interaction SIP Bulk Caller requires to register dynamically to your CIC server. If some or all stations are configured for dynamic registration with CIC Server, use a stationset to simulate registration and authentication of those stations from Interaction SIP Bulk Caller.

# Installing Interaction SIP Bulk Caller

This subject includes:

- [Upgrading Interaction SIP Bulk Caller](#)
- [Installing Interaction SIP Bulk Caller software](#)
- [Uninstalling Interaction SIP Bulk Caller](#)

## Upgrading Interaction SIP Bulk Caller

To upgrade Interaction SIP Bulk Caller:

1. Uninstall the old version of Interaction SIP Bulk Caller. For specific instructions, see [Uninstalling Interaction SIP Bulk Caller](#).
2. Install the new version. For specific instructions, see [Installing Interaction SIP Bulk Caller](#).

## Uninstalling Interaction SIP Bulk Caller

To uninstall Interaction SIP Bulk Caller:

1. In case you need them in the future, back up the files:
  - SIPBulkCallerConfig.xml
  - SampleCallSet.xml
  - SampleStationSet.xml
  - DefaultInboundCallSet.xml files

Uninstalling Interaction SIP Bulk Caller deletes those files. However, it does not delete your audio files and callset files.

2. Stop the Interaction SIP Bulk Caller service in Windows.

The name of the service is *Interaction SIP Bulk Caller*.

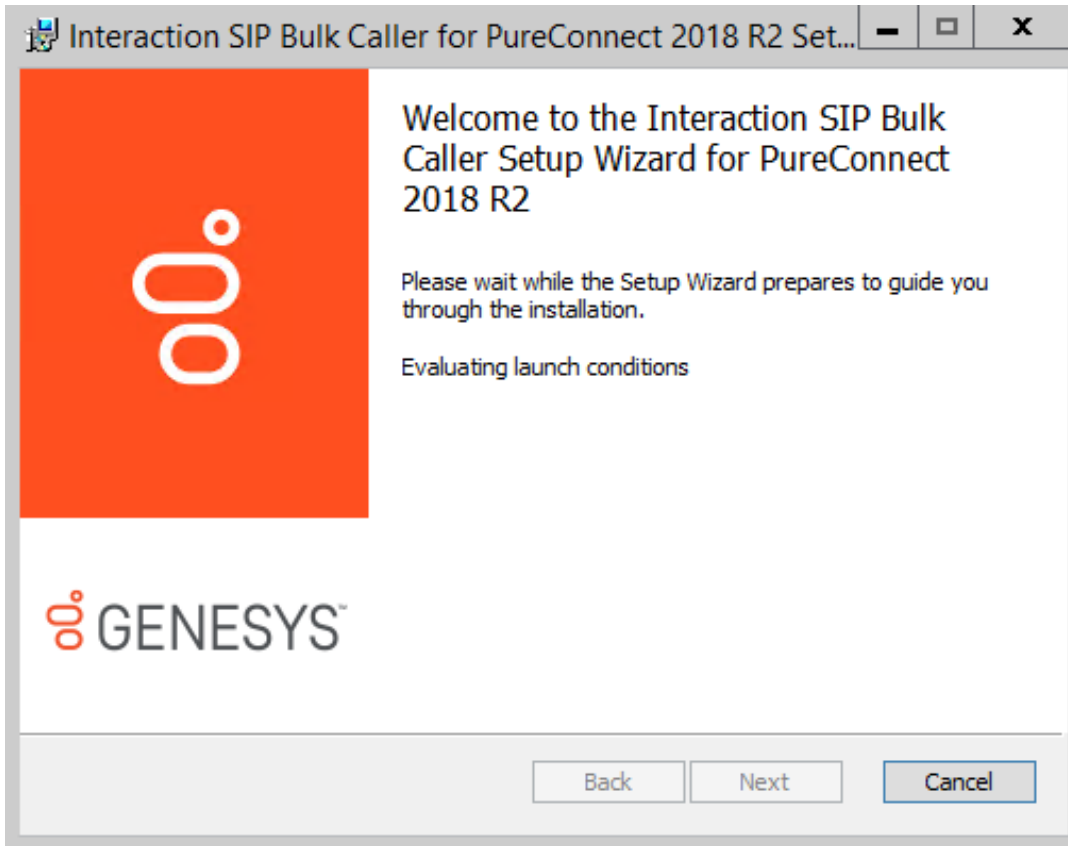
3. On the Windows Control Panel:
  - a. Start Add/Remove Programs.
  - b. Select Interaction SIP Bulk Caller.
  - c. Click Remove.

Windows uninstalls Interaction SIP Bulk Caller from your system.

## Installing Interaction SIP Bulk Caller software

To install the software:

1. Download the setup file (InteractionSIPBulkCaller.msi) from "Utilities and Downloads" on the CIC Product Information site at <https://my.inin.com/products/cic/pagees/default.aspx>
2. Browse to and double-click the setup file.



3. Work through the first two screens and then click the **Install** button.
4. Once the installation is complete, click the **Finish** button.

The setup program installs Interaction SIP Bulk Caller as a Windows service. The name of the service is *Interaction SIP Bulk Caller*.

The setup program also installs a copy of this technical reference in the following folder:

`$SBC_ROOT\UserGuide`

The setup program creates two shortcuts that link to this technical reference. The first shortcut is on the Windows Desktop. The second is on the Windows Start menu under Program Files | Genesys | Interaction SIP Bulk Caller | User Guide.

After you have installed the program, you need to apply the license.

## Applying the License

After installing Interaction SIP Bulk Caller, you must apply the Interaction SIP Bulk Caller license you obtained from Genesys. To apply the license:

1. Copy the license file to any local folder.  
Genesys recommends the folder `$SBC_ROOT\Resources`, where you installed Interaction SIP Bulk Caller.
2. Start the Interaction SIP Bulk Caller service.
3. Log on to the SIPBulkCaller web interface (<http://localhost:8080>).  
The default user name is **admin** and **1234** is the default password.
4. Select the **Configuration** tab and then select **License**.



5. Click **Choose File**, then browse to and select the license file.
6. Click **Submit** to update Interaction SIP Bulk Caller's license data.  
Interaction SIP Bulk Caller shows the license details.



## Configuration

- Outbound CallSets**
- Inbound CallSets**
- StationSets**
- Add CallSet/StationSet**
- SIPLine Config**
- Audio Config**
- License**
- Administration**

### License Configuration

<b>License Status</b>	License expires in 7.5 days.
<b>License File</b>	C:\Program Files (x86)\Interactive Intelligence\SIP Bulk Caller\Resources\SIPBulkCallerLicense_ININ_20180321081841578.xml
<b>Product</b>	I3_FEATURE_INTERNAL_SIP_BULK_CALLER
<b>Outbound CallSets</b>	Allowed
<b>Inbound CallSets</b>	Allowed
<b>StationSets</b>	Allowed
<b>Max Outbound CallSets</b>	No Limit
<b>Max Active Calls per CallSet</b>	No Limit
<b>Max CallRate</b>	No Limit
<b>Max G729 Sessions</b>	No Limit

[Add/Update License](#)

### Note:

On the Status tab, you can click **About** to view details of Interaction SIP Bulk Caller version, build date, and local machine information.

# Configuring and Using the Software

Topics covering this subject:

- [Adding a Callset or Stationset](#)
- [Editing, Reloading, or Deleting a Callset or Stationset](#)
- [Edit SIP Line Configuration](#)
- [Edit Audio Configuration](#)
- [Start or Stop Callset or Stationset](#)
- [Generating Security Certificates for TLS](#)
- [Location of Log Files](#)
- [Troubleshooting](#)



## Adding a Callset or Stationset

For inbound Callset, outbound callset, and stationset, you must create the XML configuration file. For more information, see:

- [Inbound Callset Configuration File](#)
- [Outbound Callset Configuration File](#)
- [StationSet Configuration File](#)

### Note:

Default files for inbound callset, outbound callset, and stationset are included with the installation. You can use the inbound callset file as is. You must modify the outbound callset and stationset files for your specific site characteristics.

To add a callset or stationset:

1. Create the required configuration file:
  - For an inbound callset, create the inbound callset configuration file. Put it and all audio files in the Interaction SIP Bulk Caller installation resources folder.
  - For an outbound callset, create the outbound callset configuration file. Put it and all audio files in the Interaction SIP Bulk Caller installation resources folder.
  - For a stationset, create the stationset configuration file. Put it in the Interaction SIP Bulk Caller installation resources folder.
2. Start Interaction SIP Bulk Caller, then:
  - a. Display the **Configuration** page.
  - b. Click **Add CallSet/StationSet**.

**SIP Bulk Caller**

Status Config Logout

### Configuration

- Outbound CallSets
- Inbound CallSets
- StationSets
- Add CallSet/StationSet**
- SIPLine Config
- Audio Config
- License
- Administration

1. Create a callset/stationset configuration file. If a callset is using audio files to play, then make place all referenced audio files in the resource directory (C:\Program Files (x86)\Interactive Intelligence\SIP Bulk Caller\Resources).

2. Select a set type from drop-down list.

3. Browse to your callset/stationset configuration file. Note that the file will be copied to the resource directory, and will overwrite an identically named file.

Type:

Callset File:  No file chosen

3. Select **Inbound CallSet**, **Outbound CallSet**, or **StationSet**, then input or browse to the name of the callset or stationset.
4. Click **Submit**.

Interaction SIP Bulk Caller adds your new callset or stationset.

## Editing, Reloading, or Deleting a Callset or Stationset

To edit, reload, or delete a callset or stationset:

1. Display the **Configuration** page, then click the tab for the desired type of callset or stationset.

The screenshot shows the 'SIP Bulk Caller' web interface. The 'Configuration' section is active, displaying the configuration for 'CallSet TestOutboundCallSet.xml'. The configuration parameters are as follows:

Path	C:\Program Files (x86)\Interactive Intelligence\SIP Bulk Caller\Resources
MaximumCalls	-1
ActiveCalls	80
Burst	2
InterBurstDelay	1 seconds
NoAnswerSeconds	30 seconds
MediaType	Normal
CallDurationWarning	900 seconds
MaxCallWarmDownTime	0 seconds
<b>ToAddressList</b>	Type:RoundRobin
John	sip:10001@qf-env3-cic
John1	sip:10002qf-env3-cic
<b>FromAddressList</b>	Type:RoundRobin
Henry	sip:3000@localhost
<b>Script SampleScript</b>	AudioFormat:G711 RFC2833:True
Play Action	Duration:200 seconds File:ICMainMenu.wav

Each tab displays configuration information, such as the name and path of the configuration file. It also has Edit, Delete, and Refresh buttons.

2. To make changes in the configuration file for the callset or stationset:

- a. Click **Edit**.

Interaction SIP Bulk Caller displays the XML configuration in a text editing box.

- b. Make any desired changes in the configuration.

- c. Click **Submit**.


If an error occurs, then Interaction SIP Bulk Caller displays an error message.

3. To delete the callset or stationset, click **Delete**.


Interaction SIP Bulk Caller unloads the callset or stationset. However, it does not delete the configuration file for the callset or stationset.

## Edit SIP Line Configuration

The SIP Line Configuration page shows current SIP line configuration. You may change protocol, port number and other SIP line-specific elements. Since Interaction SIP Bulk Caller has only one SIP line, you do not need to define the name of the SIP line.











# SIP Bulk Caller


Status [Config](#) [Logout](#)

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## Configuration

-  **Outbound CallSets**
-  **Inbound CallSets**
-  **StationSets**
-  **Add CallSet/StationSet**
-  **SIPLine Config**
-  **Audio Config**
-  **License**
-  **Administration**

[Refresh](#)

### SIP Line Configuration

Make changes to the SIP Line configuration. Stop all CallSets/StationSets prior to making changes in SIP line configuration.

<b>Network Interface</b>	<input type="text" value="Ethernet (172.26.23.182)"/>
<b>IP Family</b>	<input type="text" value="IPv4"/>
<b>Domain Name</b>	<input type="text" value="inin.com"/>
<b>Protocol</b>	<input type="text" value="UDP"/>
<b>Port Number</b>	<input type="text" value="5060"/>
<b>SIP QoS DSCP Value</b>	<input type="text" value="18 (24, 011000) CS3"/>
<b>TCP Connect Timer (ms)</b>	<input type="text" value="3500"/>
<b>UDP T1 Timer (ms)</b>	<input type="text" value="500"/>
<b>UDP T2 Timer (ms)</b>	<input type="text" value="4000"/>
<b>Max Packet Retry</b>	<input type="text" value="10"/>
<b>Max Invite Retry</b>	<input type="text" value="5"/>
<b>Fax Protocol</b>	<input type="text" value="T.38"/>
<b>Enable PRACK</b>	<input type="text" value="NO"/>

Root Certificate File, Line Certificate File and Line PrivateKey File are required when TLS protocol is selected. Obtain certificate files from appropriate authority. Browse to your certificate files below.

<b>Root Certificate File</b>	<input type="button" value="Choose File"/> <input type="text" value="No file chosen"/>
<b>Currently Selected File</b>	
<b>Line Certificate File</b>	<input type="button" value="Choose File"/> <input type="text" value="No file chosen"/>
<b>Currently Selected File</b>	
<b>Line PrivateKey File</b>	<input type="button" value="Choose File"/> <input type="text" value="No file chosen"/>
<b>Currently Selected File</b>	

The table explains each element of the SIP Line Configuration page.

Element	Explanation
NetworkInterface	The network connection name on your local machine. The default is Local Area Connection. If you have more than one network interface, or if the name of the network interface is different from the default, you can change the name here. Unless the correct name is set, the SIP engine will not start.
IP Family	The family of protocols used. For more information, see <a href="http://wiki.wireshark.org/InternetProtocolFamily">http://wiki.wireshark.org/InternetProtocolFamily</a> .
DomainName	The domain name for unique identification.
Protocol	The protocol to use for SIP communication. The default protocol is UDP. Available protocols are UDP, TCP, and TLS.
PortNumber	The port number to bind the SIP line. The default is 5060.
SIP QoS DSCP Value	The default DSCP value is 24 (0x18). Select the value from the drop-down list. At the time of installation, Interaction SIP Bulk Caller sets a value of 0 for registry HKEY_LOCAL_MACHINE\SYSTEM\CurrentControlSet\Services\Tcpip\Parameters\DisableUserTOSSetting. This enables Interaction SIP Bulk Caller to send the QoS byte with a SIP message. Any changes to the registry value require reboot of the machine.
TCP Connect Timer	This is the timer setting for the TCP protocol. In most cases, the default works fine.
UDP T1Timer	This is a timer setting for the UDP protocol. In most cases, the default works fine.
UDP T2Timer	This is a timer setting for the UDP protocol. In most cases, the default works fine.
Max Packet Retry	This is the maximum number of times Interaction SIP Bulk Caller will retry sending a packet. In most cases, the default works fine.
Max Invite Retry	This is the maximum number of times Interaction SIP Bulk Caller will retry sending an invitation. In most cases, the default works fine.
Fax Protocol	The fax protocol to use.
Enable PRACK	Whether or not to use PRACK as defined in RFC 3262: "Reliability of Provisional Responses in the Session Initiation Protocol (SIP)."

The following three elements apply only to the TLS (Transport Layer Security) protocol and only work if you select TLS as the SIP protocol. For more information about using TLS, see the section [Generating Security Certificates for TLS](#).

Element	Explanation
RootCertificateFile	The path for the Authority certificate file.
CertificateFile	The path for the Certificate file. This certificate must be signed by the Authority Certificate.
PrivateKeyFile	The path for the private key file. The certificate file and private key file are generated by the Authority certificate file.

## Edit Audio Configuration

The Audio Configuration page displays the current audio configuration.

**SIP Bulk Caller** Status [Config](#) [Logout](#)

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## Configuration

- Outbound CallSets**
- Inbound CallSets**
- StationSets**
- Add CallSet/StationSet**
- SIPLine Config**
- Audio Config**
- License**
- Administration**

[Refresh](#)

### Audio Configuration

Make changes to the audio configuration. Stop all CallSets/StationSets prior to making changes in audio configuration.

RTP Network Interface	Ethernet (172.26.23.182) ▼
Play Audio	Yes ▼
Audio Protocol	RTP ▼
Enable RTP QoS Tagging	Yes ▼
RTP QoS DSCP Value	2E (46, 101110) EF ▼
Enable RTCP QoS Tagging	Yes ▼
Disconnect On Broken RTP	No ▼
RTCP QoS DSCP Value	2E (46, 101110) EF ▼
Audio Engine Selection Algorithm	Dynamic ▼
Enable RTP Diagnostic Capture	No ▼
RTP Diagnostic Capture Directory	C:/windows/Temp\ININ_SBC_Diagnostics

The table explains each element of the Audio Configuration page.

Element	Explanation
RTP Network Interface	The network interface for audio data. The default is Local Area Connection. If you have a dual NIC machine, then you may set up different network interfaces for SIP data and audio data. Set the network interface for SIP data on the SIP Line Configuration page.
Play Audio	The default is Yes. If you set this to No, then Interaction SIP Bulk Caller plays no audio, it does not activate the audio source, and it ignores script configuration. This setting is useful in testing of SIP-only messages.
Audio Protocol	There are options to play either RTP or SRTP for the audio. In most situations, SRTP works well with the TLS SIP protocol.
Enable RTP QoS Tagging	If set to Yes, then RTP packets add Quality of Service (QoS) tagging.
RTP QoS DSCP Value	This is the DSCP value for RTP Quality of Service (QoS). The default value is 46 (0x2E). Select the value from the drop-down list.
Enable RTCP QoS Tagging	If set to Yes, then RTCP packets will add Quality of Service (QoS) tagging.
RTCP QoS DSCP Value	This is the DSCP value for RTCP Quality of Service (QoS). The default value is 46 (0x2E). Select the value from the drop-down list.
Audio Engine Selection Algorithm	The default algorithm is FewestGraphs, which works well in most situations. If the system has more than four processor cores and Windows Server 2008, then set the algorithm to LoadWeightedStochastic and set HLKM\System\CurrentControlSet\Services\Ndis\Parameters\MaxNumRssCpus (DWORD) to the number of CPUs you have. This balances the load across all CPUs so your system can handle more calls.
Enable RTP Diagnostic Capture	If set to Yes, then Interaction SIP Bulk Caller records the receiving audio stream for each call in HPAA Capture format. It can be used for troubleshooting.
RTP Diagnostic Capture Directory	The root directory below which to store diagnostic capture data. Beneath the root directory, Interaction SIP Bulk Caller creates a separate directory for each callset and adds the default directory path to this text box.

## Start or Stop Callset or Stationset

The Statistics page lets you start or stop callsets and Stationsets. It shows all outbound callsets, inbound callsets, and stationsets loaded into Interaction SIP Bulk Caller. It also displays callset statistics such as calls placed, received, and answered.

Once you start a CallSet, you can save a copy of the general or detailed statistics by clicking the Export button at the bottom of the page. The statistics are saved in a CSV file in the Resources folder. When the file is saved, a unique identifier will be appended to

the filename. This allows you to export the statistics file multiple times without overwriting previous export files.

**SIP Bulk Caller**

Status Config Logout

## Status

Refresh statistics automatically every 60 sec Refresh

**Statistics**

Click CallSet/StationSet name to view detailed statistics. Once CallSet is started, it shows WarmStop and HardStop button. WarmStop button stops callset, but let currently active calls to finish. HardStop button stops callset and also immediately stops currently active calls.

**Outbound CallSets**

Name	Placed	Answered	Active	Disconnected		
<a href="#">TestOutboundCallSet.xml</a>	7294	0	0	0	Start	

**Inbound CallSets**

Name	Received	Answered	Active	Disconnected		
<a href="#">DefaultInboundCallSet.xml</a>	0	0	0	0	Start	

**StationSets**

Name	Total	Registered	Requested	Accepted		
<a href="#">SampleStationSet.xml</a>	11	0	0	0	Start	
<a href="#">TestStationSet.xml</a>	1000	0	2000	2000	Start	

**MWI Statistics Details**

Type	Count	Description
Current MWI Notifications	0	The number of MWI notifications received, since the last file write.
Aggregate MWI Notifications	0	The number of MWI notifications received, since the SIP Bulk Caller service was started.
MWI On SIP Stations	0	The number of SIP stations that have the MWI On status.
MWI Off SIP Stations	0	The number of SIP stations that have the MWI Off status.
Total SIP Stations	0	The total number of MWI notifications received by the SIP Bulk Caller from the SIP stations.

Export statistics Export

To start a callset:

- Click **Start** in the callset's row.

Note: If your application also receives calls, you must start the appropriate inbound callset. If you do not start the inbound callset, Interaction SIP Bulk Caller will reject inbound calls.

After you start at least one callset, the Statistics page displays buttons to stop active callsets. There are two ways to stop a callset:

**SIP Bulk Caller**

Status Config Logout

---

## Status

Refresh statistics automatically every  Refresh

**Statistics**

Click CallSet/StationSet name to view detailed statistics. Once CallSet is started, it shows WarmStop and HardStop button. WarmStop button stops callset, but let currently active calls to finish. HardStop button stops callset and also immediately stops currently active calls.

**Outbound CallSets**

Name	Placed	Answered	Active	Disconnected		
<a href="#">TestOutboundCallSet.xml</a>	7126	0	0	0	HardStop	WarmStop

**Inbound CallSets**

Name	Received	Answered	Active	Disconnected		
<a href="#">DefaultInboundCallSet.xml</a>	0	0	0	0	Start	

**StationSets**

Name	Total	Registered	Requested	Accepted		
<a href="#">SampleStationSet.xml</a>	11	0	0	0	Start	
<a href="#">TestStationSet.xml</a>	1000	0	2000	2000	Start	

**MWI Statistics Details**

Type	Count	Description
Current MWI Notifications	0	The number of MWI notifications received, since the last file write.
Aggregate MWI Notifications	0	The number of MWI notifications received, since the SIP Bulk Caller service was started.
MWI On SIP Stations	0	The number of SIP stations that have the MWI On status.
MWI Off SIP Stations	0	The number of SIP stations that have the MWI Off status.
Total SIP Stations	0	The total number of MWI notifications received by the SIP Bulk Caller from the SIP stations.

Export statistics

- **HardStop** stops placing and receiving new calls. In addition, it immediately stops all currently placed or received calls.
- **WarmStop** stops placing and receiving new calls. Unlike HardStop, it allows currently active calls to finish.

To view detailed statistics about a callset:

1. Click the link for the callset.

Interaction SIP Bulk Caller displays detailed statistics for the callset.





## Status



Refreshing statistics automatically every 60 sec.

### Statistics

[Back](#)



### About

### Statistics Details of Outbound CallSet TestOutboundCallSet.xml

	Current	Aggregate	Description
Placed	656	7950	Calls placed by the SIP Bulk Caller. Placed = Answered + UnansweredTotal.
Answered	0	0	Calls answered by a remote party. Answered = Active + LocalDisconnected + RemoteDisconnected.
Active	0	0	Calls answered and currently playing.
LocalDisconnected	0	0	Calls answered and disconnected by the SIP Bulk Caller.
RemoteDisconnected	0	0	Calls answered and disconnected by a remote party.
AverageAnswerTime (ms)	0	0	Average time to receive SIP answer.
MaxAnswerTime (ms)	0	0	Maximum time to receive SIP answer.
AverageAnswerTime100 (ms)	0	0	Average time to receive SIP answer for the last 100 calls.
StddevAnswerTime100 (ms)	0	0	Standard deviation of SIP answer time for the last 100 calls.
AverageAudioTime (ms)	0	0	Average time to receive first RTP packet from a remote party.
MaxAudioTime (ms)	0	0	Maximum time to receive first RTP packet from a remote party.
AverageAudioTime100 (ms)	0	0	Average time to receive first RTP packet from a remote party for the last 100 calls.
StddevAudioTime100 (ms)	0	0	Standard deviation of audio time for the last 100 calls.
UnansweredTotal	656	7950	UnansweredTotal = WaitingForAnswer + Unanswered + Rejected.
WaitingForAnswer	40	40	Calls placed but not yet answered by a remote party.
Unanswered	297	3952	Calls unanswered either due to remote party not responding or local errors.
Rejected	319	3958	Calls rejected by a remote party.
LocalDisUnanswered	0	38	Calls unanswered and disconnected by the SIP Bulk Caller.
RemoteDisUnanswered	616	7872	Calls unanswered and disconnected by a remote party.
Redirected	0	0	Calls redirected to another location by a remote party.
BlindTransferred	0	0	Calls blindly transferred to another location by a remote party.

Aggregate statistics show aggregate counts since service was started or callset/stationset was modified.

Export statistics

2. Click the **Back** link at the top left of the page to return to the main page.

## Generating Security Certificates for TLS

If you select the Transport Layer Security (TLS) protocol in SIP Line configuration, then Interaction SIP Bulk Caller needs security certificate files to operate.

To generate security certificates for TLS:

1. On the CIC server, navigate to the `\I3\IC\Certificates\Lines` folder.
2. Make a backup of all current line certificates in the folder.

This step is very important! Do not skip it. You will need the backup in Step 7.

3. In Windows, open command window:
  - a. On the Windows Start menu, click **Run**.
  - b. In the Open text box, type `cmd` and then click **OK**.
4. In the command window, navigate to the `\I3\IC\Server` folder.
5. In that folder, create a new line certificate by entering:

```
GenSSLCertsU.exe -l sipbulkcalleripaddress
```

6. Copy the new line certificate file and Private Key file from the `\I3\IC\Certificates\Lines` folder to the `$SBC_ROOT\Resources` folder on the Interaction SIP Bulk Caller machine.

These files are CertificateFile and PrivateKeyFile in Interaction SIP Bulk Caller.

7. On the CIC Server, delete any new certificates and restore the original line certificate files from your backup.
8. Copy the **ServerGroupCertificate.cer** file from `\I3\IC\Certificates\ServerGroup` folder to the `$SBC_ROOT\Resources` folder on the Interaction SIP Bulk Caller machine.

This would be RootCertificateFile in Interaction SIP Bulk Caller.

9. Start Interaction SIP Bulk Caller and use the [SIP Line Configuration page](#) to make any required changes.

## Location of Log Files

Interaction SIP Bulk Caller uses same tracelib to write logs as do other Customer Interaction Center components. Logs are in the `%ININ_TRACE_ROOT%\inin_tracing` directory.

You need to define the **ININ\_TRACE\_ROOT** environment variable and provide a directory where you want to write logs. If this environment variable is not defined, then Interaction SIP Bulk Caller writes logs in the `%temp%\inin_tracing` directory. You need IninTraceViewer to view the logs.

## Troubleshooting

Problem	SIP Line fails to activate.
Solution	The Network Interface parameter in SIP line configuration must match the name of the network card on the machine. The default value is Local Area Connection. If your network card has a different name or you have more than one network card on the machine, then provide the exact name and submit the changes.  Make sure some other application is not using the same port specified in the SIP line configuration.
Problem	Callset does not load.
Solution	Check the format of the audio file. AudioLib supports 8-bit 8 KHz CCITT u-law, 8-bit 8 KHz CCITT a-law, 8-bit 8 KHz GSM, and 16-bit linear PCM for audio wav files. If your audio wav file is not in one of these formats, then you must convert it before using it in Interaction SIP Bulk Caller. You can use the Windows Sounds Recorder utility (located in Program   Accessories   Entertainment) to convert the file format.  Element names and attribute names in the XML configuration file must match as expected. The XML parser is case-sensitive.
Problem	SIP Bulk Caller rejects a callset, stationset, license, or certificate file.
Solution	SIP Bulk Caller accepts alphanumeric, hyphen (-), underscore (_), and period (.) characters for callset, stationset, license and certificate files. It rejects any other file names.

# Appendix: Configuration Files

This appendix has a section for each of the configuration files you use with Interaction SIP Bulk Caller.

- [Outbound Callset Configuration File](#)
- [Inbound Callset Configuration File](#)
- [StationSet Configuration File](#)
- [Main Xml Configuration File](#)
- [Registering a station based on a SIP name](#)

## Outbound Callset Configuration File

Each callset requires a separate XML configuration file. For example, if your callset name is **ICServerTest**, then the callset XML file name must be **ICServerTest.xml**. The callset XML configuration file and audio files the callset uses must be either in the `$(SBC_ROOT)\Resources` folder if **DirPath** is not specified, or must be in the folder specified by the **DirPath** attribute.

This configuration file illustrates all options that are available. You do not need to use all the options.

For more information, see [Explanation of Outbound Callset Configuration File](#).

```
<?xml version="1.0"?>
  <CallSetConfig>
    <MaximumCalls>-1</MaximumCalls>
    <ActiveCalls>100</ActiveCalls>
    <Burst>2</Burst>
    <InterBurstDelay>2</InterBurstDelay>
    <MaxCallWarmDownTime>0</MaxCallWarmDownTime>
    <CallDurationWarning>900</CallDurationWarning>
    <MediaType>Normal</MediaType>
    <NoAnswerSeconds>30</NoAnswerSeconds>
    <AutoStart>False</AutoStart>
    <ToAddressList Type="Random">
      <Address CallerId="John">sip:1111@icserver1</Address>
      <Address CallerId="James">sip:2222@icserver1</Address>
      <AddressRange CallerId="Nancy" StartNumber="5001"
        EndNumber="5100" MachineAddr="icserver1"/>
    </ToAddressList>
    <FromAddressList Type="RoundRobin">
      <Address CallerId="Joe Smith">sip:anyext@localmachinename</Address>
    </FromAddressList>
    <Script Name="SampleScript" AudioFormat="G711" RFC2833Dtmf="True" AbsoluteCallLength="0"
      CheckAudioFiles="True">
      <Action Type="Wait" Duration="15"/>
      <Action Type="Dtmf" Digits="34#" InterDigitDelay="50" ToneDuration="100"/>
      <Action Type="Wait" Duration="5"/>
      <Action Type="SequenceDtmf" StartDigits="1000" EndDigits="2000" InterDigitDelay="50"
        ToneDuration="100"/>
      <Action Type="RandomDtmf" StartDigits="1000" EndDigits="2000" InterDigitDelay="50" ToneDuration="100"/>
      <Action Type="Play" FileName="Yourturn.wav"/>
      <Action Type="Play" Duration="15" FileName="ICMainMenu.wav"/>
      <Action Type="RandomPlay" Duration="30">
        <FileName>AudioFile1.wav</FileName>
        <FileName>AudioFile2.wav</FileName>
        <FileName>AudioFile3.wav</FileName>
        <FileName>AudioFile4.wav</FileName>
      </Action>
      <Action Type="RandomPlay">
        <FileName>AudioFile1.wav</FileName>
        <FileName>AudioFile2.wav</FileName>
        <FileName>AudioFile3.wav</FileName>
        <FileName>AudioFile4.wav</FileName>
      </Action>
      <Action Type="SequencePlay" Duration="30">
        <FileName>AudioFile1.wav</FileName>
        <FileName>AudioFile2.wav</FileName>
        <FileName>AudioFile3.wav</FileName>
        <FileName>AudioFile4.wav</FileName>
      </Action>
      <Action Type="SequencePlay">
        <FileName>AudioFile1.wav</FileName>
        <FileName>AudioFile2.wav</FileName>
        <FileName>AudioFile3.wav</FileName>
        <FileName>AudioFile4.wav</FileName>
      </Action>
    </Script>
  </CallSetConfig>
```

---

## Explanation of Outbound Callset Configuration File

Element	Explanation
MaximumCalls	The callset stops after making the number of calls set in MaximumCalls. If the value of MaximumCalls is -1, then the callset runs continuously until the user stops it.
ActiveCalls	The number of calls that can be active simultaneously. If Interaction SIP Bulk Caller reaches this limit, then it stops placing new calls until a call is disconnected.
Burst	The total number of new calls placed together as a burst.
BurstRange	<p>If you want to set up a variable burst rate, then use the BurstRange element instead of Burst. For example,</p> <pre>&lt;BurstRange Type="Sequence" StartNumber="2" EndNumber="20" IncrStep="1"/&gt;</pre> <p>This code specifies the range of burst and the size of the increment step. If incrstep is set to 2, then burst would be 2,4,6,...,20. If type is Sequence, then the first burst of calls would be 2, the next would be 4, and so on in the cycle. If the type is Random, then the burst could have any value in the range of 2,4,6, ... 20.</p>
InterBurstDelay	<p>The delay in seconds between two bursts of calls.</p> <p>For example, if Burst=6 and InterBurstDelay=2, then Interaction SIP Bulk Caller places six calls every two seconds. You can also use decimal values in increments of 0.5, such as 0.5, 1, 1.5, and 2. For example, if Burst=3 and InterBurstDelay=0.5, then Interaction SIP Bulk Caller places three calls every 0.5 seconds.</p> <p>Remember that even if you have a high burst rate, Interaction SIP Bulk Caller stops placing new calls after it reaches the ActiveCalls limit until old calls get disconnected.</p>
MaxCallWarmDownTime	<p>This value is in seconds and the default value is 0. Use this parameter to insert a delay when outbound callset reaches its activecall limit and then a call gets disconnected. Before placing new call, Interaction SIP Bulk Caller waits for the number of seconds that this value specifies.</p> <p>This is used most often when calls are placed to a PBX or gateway where a delay is required to clear the channel before placing a new call.</p>
CallDurationWarning	This value is in seconds and the default value is 900 seconds (15 minutes). If call duration exceeds this value, then Interaction SIP Bulk Caller stores it as a warning message in its logs. This is used to track hung calls.
MediaType	The MediaType values are Normal and Delayed. In a delayed media type, media information is sent in ACK. The default value is Normal.
NoAnswerSeconds	This is the number of seconds for the sipengine will wait for a response. If a response is not received in the specified time, then the sipengine sends a disconnect message for the call. The default value is 30 seconds.
AutoStart	This is a Boolean (true or false) value. If this is set to true, then the callset starts automatically when the Interaction SIP Bulk Caller service starts.
ToAddressList	This is a list of addresses where you want to send SIP calls. Interaction SIP Bulk Caller sends calls to the addresses in either RoundRobin or Random mode based on the Type attribute value. The default is RoundRobin.
Address	This contains CallerIds and SIP addresses where Interaction SIP Bulk Caller can send SIP calls.
AddressRange	You can use this element with a range of addresses in sequence. For example, SIP addresses might be added as sip:5001@icserver1, sip:5002@icserver1, sip:5003@icserver1, ... upto sip:5100@icserver1.
FromAddressList	This is a list of "from" addresses.
Script	The Script consists of one or more Actions to execute in the SIP call.

AudioFormat	<p>The format of the audio data to play. Acceptable values are G711, G711A, G722, G726, G729AB, G729A, or GSM.</p> <p>Note that current Audiolib (HPAA) supports 8-bit 8 KHz CCITT u-law, 8-bit 8 KHz CCITT a-law, 8-bit 8 KHz GSM, and 16-bit linear PCM for audio wav files. If your audio wav file is not in one of these formats, then manually convert it before using it in Interaction SIP Bulk Caller.</p>
RFC2833Dtmf	This is a Boolean (true or false) value. If it is set to true, then Interaction SIP Bulk Caller sends an RFC2833 event for Dtmf tones. If it is set to false, then Interaction SIP Bulk Caller sends inband dtmf.
AbsoluteCallLength	If this value is 0, then Interaction SIP Bulk Caller ignores it. If it is greater than 0, Interaction SIP Bulk Caller cuts off the call after the number of seconds specified in AbsoluteCallLength. This parameter ensures that the length of SIP calls will not exceed the time specified in AbsoluteCallLength even if calls are transferred from station to station.
CheckAudioFiles	When you load the callset, Interaction SIP Bulk Caller checks the audio file using the callset. If you set this value to false, then Interaction SIP Bulk Caller does not do this check.
SendPTTime	<p>Packetization interval in milliseconds of audio sent by the script. Valid values are 10 - 160. For G729A and G729AB, the value is rounded up to the next multiple of 10 ms. This setting has no effect on:</p> <ul style="list-style-type: none"> <li>• The G722, GSM, and G723 AudioFormats.</li> <li>• The packetization interval of RFC 2833 events or T.38 UDPTL streams.</li> </ul> <p>With or without this setting, SIP Bulk Caller ignores any a=ptime and a=maxptime lines in the SDP it receives.</p>
Wait Action	This specifies the number of seconds that a call will send silence RTP packets.
Dtmf Action	Each call plays the dtmf digits specified in the Digits attribute. There is a delay of InterDigitDelay milliseconds between two dtmf digit play. You can also set the duration in milliseconds of each tone.
SequenceDtmf Action	Each call plays different dtmf digits in sequence from StartDigits to EndDigits. For example, the first call might play Dtmf 1000, the second call might play Dtmf 1001, and so on up to Dtmf 2000. After reaching EndDigits Dtmf, the cycle repeats.
RandomDtmf Action	Each call plays different random dtmf digits between StartDigits and EndDigits. For example, a call might play any dtmf digits between 1000 and 2000.
Play Action	Each call plays an audio wav file. In the FileName attribute, put only the name of the audio file; do not include the directory path. If the Play action has a Duration attribute, then a call will play audio file up to the duration.
RandomPlay Action	Each call randomly selects one file from files listed in this action and plays its full length. If the RandomPlay action has a Duration attribute, then the call will play the audio file until the end of the specified duration.
SequencePlay Action	Each call selects an audio file in sequence and plays its full length. The first call plays the first audio file, the second call plays the second audio file, and so on. If the SequencePlay action has a Duration attribute, then calls play the audio file until the end of the specified duration.
Fax Action	<p>This lets you specify the outbound calls as faxes. Interaction SIP Bulk Caller supports T.30 and T.38 fax transmission. To send a fax, add this code to the configuration file:</p> <pre>&lt;Action Type="Fax" Direction="Send" FileName="FaxSend.tif"/&gt;</pre> <p>Note that a .TIFF file (same as the FileName attribute) must be present in the resource directory for fax send. In the example, the file FaxSend.tif must be in the resource directory. TIFF files generated by the CIC Fax Client are not supported at time of release by the MediaServer Audio library that SIPBulkCaller uses.</p>
Hold Action	Each call can be placed on hold. In the Filename attribute, put only the name of the audio file. If the Hold action has a Duration attribute, then the hold lasts for the specified duration. The Hold makes the SBC send a SIP UPDATE message to the remote user agent (UA). This action simulates pressing a Hold button on an IP phone. If the Filename attribute is empty (""), the SBC sends the unspecified IP (0.0.0.0 or ::) address in the SIP UPDATE message depending on the IP family SIP Line config setting.

## Inbound Callset Configuration File

The inbound callset has fewer parameters than the outbound callset.

For more information, see [Explanation of Inbound Callset Configuration File](#).

```
<?xml version="1.0"?>
<CallSetConfig>

<StationList>
<StationRange StartExt="7000" EndExt="8000"/>
</StationList>
<WaitBeforeAnswer>0</WaitBeforeAnswer>
<CallDurationWarning>900</CallDurationWarning>
<MediaType>Normal</MediaType>
<AutoStart>False</AutoStart>

<!-- //comment
Modify script in defaultinboundcallset according to your requirement
-->
<Script Name="SampleScript" AudioFormat="G711" RFC2833Dtmf="True" AbsoluteCallLength="0"
CheckAudioFiles="True">
<Action Type="Wait" Duration="15"/>
<Action Type="Play" Duration="20" FileName="ICMainMenu.wav"/>
</Script>

</CallSetConfig>
```

---

## Explanation of Inbound Callset Configuration File

Element	Explanation
StationList	<p>This contains StationRange elements. When a call is received, it contains the station extension. Inbound callset, which contains this specific station extension, answers the call. If none of the inbound callsets contain the station extension received, then DefaultInboundCallSet answers the call. DefaultInboundCallSet does not contain the StationList element, since it answers calls for all station extensions.</p> <p>In addition to the StationRange elements, the StationList element can contain the Station Name attribute. See <a href="#">Registering a station based on a SIP name</a>.</p>
WaitBeforeAnswer	<p>This is the delay in seconds before an inbound call is answered (picked up). The default value is 0.</p> <p>This is different from a wait action. This element delays answering of the inbound call. In wait action, by contrast, a call is answered and connected, but does not play audio until after the duration specified in wait action.</p>
WaitBeforeAnswerRange	<p>If you want variable delay before answering the call, then use the WaitBeforeAnswerRange element. For example,</p> <pre>&lt;WaitBeforeAnswerRange Type="Random" StartNumber="2" EndNumber="30" IncrStep="2" /&gt;</pre> <p>Here, it creates a range of values between 2 and 30 with increments of 2. Values are 2,4,6....30. Interaction SIP Bulk Caller assigns a random value within the range to each call.</p>
CallDurationWarning	<p>This value is in seconds and the default value is 900 seconds (15 minutes). If call duration exceeds this value, then Interaction SIP Bulk Caller stores it as a warning message in its logs. This is used to track hung calls.</p>
MediaType	<p>The MediaType value can be Normal or Early. If media type is set to early, then media information is sent in Session Progress (183). In such a case, set the number of seconds in WaitBeforeAnswer. Interaction SIP Bulk Caller sends an answer after the number of seconds in WaitBeforeAnswer.</p>
AutoStart	<p>If this value is set to true, then the callset is automatically starts when the Interaction SIP Bulk Caller service starts.</p>
Script	<p>This is a set of actions executed when an inbound call is received. It is the same as the outbound callset. Refer to <a href="#">Outbound CallSet</a> for more details.</p>
Fax Action	<p>This lets you specify inbound calls as faxes. Interaction SIP Bulk Caller supports T.30 and T.38 fax transmission. To receive faxes, add this code to the configuration file:</p> <pre>&lt;Action Type="Fax" Direction="Receive" ReceiveDirectory="d:\Temp\" /&gt;</pre> <p>In the code line's <code>ReceiveDirectory</code> parameter, replace <code>d:\Temp\</code> in the example code line with the wanted directory path.</p> <p>Note that a call set can contain only one fax action. Combining a fax action with other actions (e.g., play, wait, etc.) or another fax action is not allowed.</p>
Hold Action	<p>Each call can be placed on hold. In the Filename attribute, put only the name of the audio file. If the Hold action has a Duration attribute, then the hold lasts for the specified duration. The Hold makes the SBC send a SIP UPDATE message to the remote user agent (UA). This action simulates pressing a Hold button on an IP phone. If the Filename attribute is empty (""), the SBC sends the unspecified IP (0.0.0.0 or ::) address in the SIP UPDATE message depending on the IP family SIP Line config setting.</p>
SendPTTime	<p>Packetization interval in milliseconds of audio sent by the script. Valid values are 10 - 160. For G729A and G729AB, the value is rounded up to the next multiple of 10 ms. This setting has no effect on:</p> <ul style="list-style-type: none"> <li>The G722, GSM, and G723 AudioFormats.</li> <li>The packetization interval of RFC 2833 events or T.38 UDPTL streams.</li> </ul> <p>With or without this setting, SIP Bulk Caller ignores any <code>a=ptime</code> and <code>a=maxptime</code> lines in the SDP it receives.</p>

**DefaultInboundCallSet:** DefaultInboundCallSet is installed with Interaction SIP Bulk Caller. As its name suggests, it is the default callset for inbound answering. When you have only DefaultInboundCallSet listed, then it answers calls received for all stations. If you have more than one inbound callset, then Interaction SIP Bulk Caller finds an inbound callset matching the station extension received in the inbound call. If none of the inbound callsets match the station extension, then it is answered by DefaultInboundCallSet.



## StationSet Configuration File

StationSet is used to simulate station registration from Interaction SIP Bulk Caller. It sends registration and authentication requests to the CIC server. The SIP station on the CIC server is dynamically registered from Interaction SIP Bulk Caller (sipbulkreceiver). Each stationset requires a separate XML configuration file.

The name of the XML file must be *stationsetname.xml*. SampleStationSet.xml is included as part of the installation. If you wish, you may simply edit that file.

You can use a stationset alone to test registration functionality of the CIC server. You might also need to configure and use a stationset if you have stations on CIC server configured for dynamic registration and you want to receive calls. In this scenario, configure InboundCallSet to answer the call and configure StationSet for station registration.

For more information, see [Explanation of StationSet Configuration File](#).

```
<?xml version="1.0"?>
<StationSetConfig>

<Burst>20</Burst>
<InterBurstDelay>1</InterBurstDelay>
<ExpirationInterval>3600</ExpirationInterval>
<Username></Username>
<Password></Password>

<ICServerList>
<ICServer>ICServerIPAddr</ICServer>
</ICServerList>

<StationList>
<StationRange StartExt="7000" EndExt="7100"/>
<StationRange StartExt="9500" EndExt="9700"/>
</StationList>

</StationSetConfig>
```

## Explanation of StationSet Configuration File

Element	Explanation
Burst	Total number of stations to send registration request simultaneously.
InterBurstDelay	Delay in seconds between two bursts of registrations. You can also use decimal values in increments of 0.5, such as 0.5, 1, 1.5, and 2.
ExpirationInterval	Time in seconds when registration expires for an SIP station. Interaction SIP Bulk Caller re-registers an SIP station before the SIP station registration expires. Set expirationinterval long enough to register all stations before registration of the first station expires. For example, if you have 5000 stations to register and the rate is 20 stations per second, then expirationinterval must be higher than 250 ( $5000/20 = 250$ ) seconds in order to finish registration of all stations before the first station registration expires.
Username and Password	Set username and password if an SIP Station on CIC server requires authentication at time of registration. It must be same for all SIP stations in the stationset.
ICServerList	Provide the name or IP address of the CIC server where you want your station to register. If you have a switchover pair, then add the primary and backup server names in the list.
StationList	This contains a list of StationRange elements.  In addition to the StationRange elements, the StationList element can contain the Station Name attribute. See <a href="#">Registering a station based on a SIP name</a> .
StationRange	This has StartExt and EndExt attributes. Interaction SIP Bulk Caller creates a list of station ranges from StartExt to EndExt.

## Main Xml Configuration File

The main XML configuration file is in the `$SBC_ROOT\Resources` directory. Its name must be `SIPBulkCallerConfig.xml`. You make most of the changes via the Web GUI. The elements that require manual changes are explained in [Explanation of Main Xml Configuration File](#).

```
<?xml version="1.0"?>
  <SIPBulkCallerConfig version="1.0">
    <ApplicationConfig>
      <SIPBulkCallerName></SIPBulkCallerName>
      <HttpPort>8080</HttpPort>
      <TelnetPort>0</TelnetPort>
      <MaxThreadsInPool>100</MaxThreadsInPool>
      <PlayAudio>True</PlayAudio>
      <RtpNetworkInterface>Local Area Connection</RtpNetworkInterface>
      <UseSrtp>False</UseSrtp>
      <RtpQosTagging>False</RtpQosTagging>
      <RtpQosDscpValue>46</RtpQosDscpValue>
      <RtcpQosTagging>False</RtcpQosTagging>
      <RtcpQosDscpValue>46</RtcpQosDscpValue>
      <AudioEngineSelectionAlgorithm>FewestGraphs</AudioEngineSelectionAlgorithm>
      <RtpDiagCaptureEnabled>False</RtpDiagCaptureEnabled>
      <RtpDiagCaptureDirectory></RtpDiagCaptureDirectory>
    </ApplicationConfig>

    <SIPLineConfig>
      <DomainName>inin.com</DomainName>
      <Protocol>UDP</Protocol>
      <NetworkInterface>Local Area Connection</NetworkInterface>
      <PortNumber>5060</PortNumber>
      <SIPQosByte>96</SIPQosByte>
      <T1Timer>500</T1Timer>
      <T2Timer>4000</T2Timer>
      <TCPConnectTimer>3500</TCPConnectTimer>
      <MaxPacketRetry>10</MaxPacketRetry>
      <MaxInviteRetry>6</MaxInviteRetry>
      <RootCertificateFile></RootCertificateFile>
      <CertificateFile></CertificateFile>
      <PrivateKeyFile></PrivateKeyFile>
    </SIPLineConfig>
    <!-- //comment IMP: Do not use space in callsetname. -->
    <OutboundConfig>
      <CallSet Name="SampleCallSet"/>
      <CallSet Name="ICServer1Test" DirPath="D:\MyCallSetFiles"/>
    </OutboundConfig>

    <!-- //comment IMP: Do not remove or rename DefaultInboundCallSet -->
    <InboundConfig>
      <CallSet Name="DefaultInboundCallSet"/>
      <CallSet Name="User1CallSet" DirPath="D:\MyCallSetFiles"/>
    </InboundConfig>
    <StationConfig>
      <StationSet Name="SampleStationSet"/>
    </StationConfig>
  </SIPBulkCallerConfig>
```

---

## Explanation of Main Xml Configuration File

Element	Explanation
SIPBulkCallerName	Optional. Set this value if you want to give a customized name to Interaction SIP Bulk Caller.
HttpPort	The Web interface of Interaction SIP Bulk Caller starts at port number 8080. The Web address for the application is http://localhost:8080. To start the Web interface on some other port, change port number in HttpPort.
TelnetPort	<p>The port on which Interaction SIP Bulk Caller would start the Telnet server. The default port value is 0 (zero) so by default, Interaction SIP Bulk Caller does not start the Telnet server. When this is set to 23, you can Telnet to Interaction SIP Bulk Caller. The Telnet server gives command-line access to Interaction SIP Bulk Caller. It lets you start or stop callsets and even add or delete callsets.</p> <p>The Telnet server is useful for programmatic access or batch script access to Interaction SIP Bulk Caller. You can also use it for scheduling the start and stop time of your callsets, which gives you better control of your simulation.</p>
MaxThreadsInPool	Total number of threads spawned for handling calls. You might not need to change its default value.

## Registering a station based on a SIP name

In addition to registering station extensions using the StartExt and EndExt attributes, you can now register individual stations with the SIP name by using the Station Name attribute inside the Station List tag.

### Example

```
<StationList>
<StationRange StartExt="7000" EndExt="7010"/>
<Station Name="test_name"/>
</StationList>
```

This specifies an extension range as well as a name of "test\_name". This new tag can be used in incoming call sets (to selectively answer calls based on incoming name) as well as station sets (for registration).

## Change Log

Date	Changes
28-July-2010	Initial release of this document.
18-August-2010	In section on installation, added setup file location on the Support Site.
01-September-2010	Updated figures to show user interface changes. Made minor changes in procedures and text, such as changing the name of the Windows Service.
26-October-2012	Added information to the appendix about fax support in callsets; licensing page changes; browsing to configuration files.
08-April-2013	Added information about T.30 fax support. Updated copyrights and made other minor tweaks.
03-April-2014	Updated information about SIP line configuration, audio configuration, and other sections. Updated copyrights.
20-September-2014	Updated documentation to reflect changes required in the transition from version 4.0 SU# to CIC 2015 R1, such as updates to product version numbers, system requirements, installation procedures, references to Interactive Intelligence Product Information site URLs, and copyright and trademark information.
07-October-2014	Updated the SIP Line Config page figure and accompanying table in the section "Edit SIP Line Configuration" to include IPv6 and PRACK, along with other minor changes. Added coverage of the new Hold action to the config tables for inbound and outbound callsets.
13-October-2014	Added row about file name requirements to table in Troubleshooting. Added coverage of the new script attribute SendPTIME to the sections on Outbound and Inbound callsets.
12-June-2015	Updated cover page to reflect new color scheme and logo. Updated copyright and trademark information.
09-October-2015	<ul style="list-style-type: none"> <li>• Updated MSXML Parser version from 4.0 to 6.0 in System Requirements.</li> <li>• Added information about a new feature that allows you to export callset statistics from the Status &gt; Statistics page.</li> <li>• Updated screen shots</li> </ul>
04-February-2016	Updated documentation to reflect 2016 R2 Release
06-April-2016	<ul style="list-style-type: none"> <li>• Updated documentation to reflect 2016 R3 Release</li> <li>• Updated copyright and trademark information.</li> <li>• Added information about the new Station Name attribute, which allows you to register a station with a SIP name.</li> </ul>
05-February-2018	Conversion to HTML.
21-March-2018	Rebranded screen captures and terminology.
18-July-2019	Reorganized the content only, which included combining some topics and deleting others that just had an introductory sentence such as, "In this section...".

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